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EXAMINER	
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ART UNIT	PAPER NUMBER
2644	

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Please find below and/or attached an Office communication concerning this application or proceeding.

WH

Office Action Summary	Application No.	Applicant(s)	
	09/473,547	BENESTY ET AL.	
	Examiner	**Art Unit**	
	Tony M. Jacobson	2644	

-- The MAILING DATE of this communication appears on the cover sheet with the correspondence address --

Period for Reply

A SHORTENED STATUTORY PERIOD FOR REPLY IS SET TO EXPIRE 3 MONTH(S) FROM THE MAILING DATE OF THIS COMMUNICATION.

- Extensions of time may be available under the provisions of 37 CFR 1.136(a). In no event, however, may a reply be timely filed after SIX (6) MONTHS from the mailing date of this communication.
- If the period for reply specified above is less than thirty (30) days, a reply within the statutory minimum of thirty (30) days will be considered timely.
- If NO period for reply is specified above, the maximum statutory period will apply and will expire SIX (6) MONTHS from the mailing date of this communication.
- Failure to reply within the set or extended period for reply will, by statute, cause the application to become ABANDONED (35 U.S.C. § 133). Any reply received by the Office later than three months after the mailing date of this communication, even if timely filed, may reduce any earned patent term adjustment. See 37 CFR 1.704(b).

Status

1)☒ Responsive to communication(s) filed on 28 December 1999.

2a)☐ This action is **FINAL**. 2b)☒ This action is non-final.

3)☐ Since this application is in condition for allowance except for formal matters, prosecution as to the merits is closed in accordance with the practice under *Ex parte Quayle*, 1935 C.D. 11, 453 O.G. 213.

Disposition of Claims

4)☒ Claim(s) 1-52 is/are pending in the application.

4a) Of the above claim(s) _____ is/are withdrawn from consideration.

5)☐ Claim(s) _____ is/are allowed.

6)☒ Claim(s) 1-52 is/are rejected.

7)☐ Claim(s) _____ is/are objected to.

8)☐ Claim(s) _____ are subject to restriction and/or election requirement.

Application Papers

9)☐ The specification is objected to by the Examiner.

10)☒ The drawing(s) filed on 27 April 2000 is/are: a)☒ accepted or b)☐ objected to by the Examiner.

Applicant may not request that any objection to the drawing(s) be held in abeyance. See 37 CFR 1.85(a).

Replacement drawing sheet(s) including the correction is required if the drawing(s) is objected to. See 37 CFR 1.121(d).

11)☐ The oath or declaration is objected to by the Examiner. Note the attached Office Action or form PTO-152.

Priority under 35 U.S.C. § 119

12)☐ Acknowledgment is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d) or (f).

a)☐ All b)☐ Some * c)☐ None of:

1.☐ Certified copies of the priority documents have been received.

2.☐ Certified copies of the priority documents have been received in Application No. _____.

3.☐ Copies of the certified copies of the priority documents have been received in this National Stage application from the International Bureau (PCT Rule 17.2(a)).

* See the attached detailed Office action for a list of the certified copies not received.

Attachment(s)

1) ☒ Notice of References Cited (PTO-892)

2) ☐ Notice of Draftsperson's Patent Drawing Review (PTO-948)

3) ☒ Information Disclosure Statement(s) (PTO-1449 or PTO/SB/08) Paper No(s)/Mail Date 12/28/99

4) ☐ Interview Summary (PTO-413) Paper No(s)/Mail Date. _____.

5) ☐ Notice of Informal Patent Application (PTO-152)

6) ☐ Other: _____.

DETAILED ACTION

Specification

1. The disclosure is objected to because of the following informalities: The fourth equation following Equation 7 at page 22 of the specification appears to contain a typographical error; it appears Applicants intended to recite $\hat{\underline{y}}(m) = F\hat{y}'(m)$, instead of $\hat{y}'(m) = F\hat{y}'(m)$ in accordance with the symbol convention stated at page 18, lines 13-17.

Appropriate correction is required.

Claim Objections

2. **Claims 8-10** are objected to under 37 CFR 1.75 as being a substantial duplicate of claims 3-5. When two claims in an application are duplicates or else are so close in content that they both cover the same thing, despite a slight difference in wording, it is proper after allowing one claim to object to the other as being a substantial duplicate of the allowed claim. See MPEP § 706.03(k). It appears that claims 8-10 may have been intended to depend upon claim 6, rather than claim 1, thus forming in claims 6-10 a structure equivalent to that of claims 1-5. The following prior-art rejections are based on this assumption. If in fact Applicants intended claims 8-10 to depend on claim 1, they can be considered rejected on the same grounds as claims 3-5, since they exactly duplicate those claims.

3. **Claim 32** is objected to under 37 CFR 1.75(c), as being of improper dependent form for failing to further limit the subject matter of a previous claim. Applicant is required to cancel the claim(s), or amend the claim(s) to place the claim(s) in proper dependent form, or rewrite the claim(s) in independent form. The set of equations recited in claim 32 (apparently indicating a constrained algorithm) seem to be mutually exclusive form those of claim 30 (apparently indicating an unconstrained algorithm), upon which claim 32 depends; i.e., Applicant has not disclosed an embodiment in which the constrained and unconstrained algorithms are both applied concurrently. Because a proper dependent claim must contain all the limitations of a previous claim to which it refers, while further limiting the subject matter of the previous claim, claim 32 is not a proper dependent claim, as it cannot contain all the limitations of claim 30.

Claim Rejections - 35 USC § 112

4. The following is a quotation of the first paragraph of 35 U.S.C. 112:

> The specification shall contain a written description of the invention, and of the manner and process of making and using it, in such full, clear, concise, and exact terms as to enable any person skilled in the art to which it pertains, or with which it is most nearly connected, to make and use the same and shall set forth the best mode contemplated by the inventor of carrying out his invention.

5. **Claims 2, 7, 12, 19, 27, 34, 39, and 47** are rejected under 35 U.S.C. 112, first paragraph, because the specification, while being (marginally) enabling for forming a circulant matrix by augmentation of a matrix of vectors representing an input signal, does not reasonably provide enablement for forming a circulant matrix by augmenting the input signal in general (e.g., by adding another signal to the input signal or

amplifying the input signal). The specification does not enable any person skilled in the art to which it pertains, or with which it is most nearly connected, to make and/or use the invention commensurate in scope with these claims.

6. **Claims 2, 3, 7, 8, 12, 13, 19, 20, 27, 28, 34, 35, 39, 40, 47, and 48** are rejected under 35 U.S.C. 112, first paragraph, as based on a disclosure which is not enabling. Recursively calculating the inverse of a power spectral density ($\boldsymbol{S}$) of the diagonal vector ($\boldsymbol{D}$); calculating a cross spectrum between the Fourier transform of an error signal ($\underline{\mathbf{e}}$) and the complex conjugate or complex conjugate transpose of the diagonal matrix ($\boldsymbol{D}$); forming a product between the inverse of the power spectral density ($\boldsymbol{S}^{-1}$), the complex conjugate (transpose) of the diagonal matrix ($\boldsymbol{D}^{\mathbf{H}}$), the Fourier transform of the error signal ($\underline{\mathbf{e}}$), and a constant (μ_u or $[1-\lambda_f]$); and recursively updating the estimate by adding the product so determined to the previous value of the estimate (or an equivalent description of the steps disclosed), critical or essential to the practice of the invention, but not included in the claim(s) is not enabled by the disclosure. See *In re Mayhew*, 527 F.2d 1229, 188 USPQ 356 (CCPA 1976). Applicants' have not disclosed an embodiment of the invention in which diagonally decomposing an augmented input matrix by Fourier transformation alone produces a useable estimate of the impulse response of an echo path. (See MPEP 2172.01.)

7. **Claims 26 and 33** are rejected under 35 U.S.C. 112, first paragraph, as based on a disclosure which is not enabling. Subtracting said [echo signal] estimates from the

signal transmitted over the [channel from said second location to said first location], critical or essential to the practice of the invention, but not included in the claim(s) is not enabled by the disclosure. See *In re Mayhew*, 527 F.2d 1229, 188 USPQ 356 (CCPA 1976). The disclosure generally indicates that the echo/"impulse response" estimate must be subtracted form the transmitted signal in order to achieve distortion (echo) cancellation, and does not disclose an embodiment in which the desired result is achieved without execution of such a step.

8. The following is a quotation of the second paragraph of 35 U.S.C. 112:

> The specification shall conclude with one or more claims particularly pointing out and distinctly claiming the subject matter which the applicant regards as his invention.

9. **Claims 1-52** are rejected under 35 U.S.C. 112, second paragraph, as being indefinite for failing to particularly point out and distinctly claim the subject matter which applicant regards as the invention.

10. Where applicant acts as his or her own lexicographer to specifically define a term of a claim contrary to its ordinary meaning, the written description must clearly redefine the claim term and set forth the uncommon definition so as to put one reasonably skilled in the art on notice that the applicant intended to so redefine that claim term. *Process Control Corp. v. HydReclaim Corp.*, 190 F.3d 1350, 1357, 52 USPQ2d 1029, 1033 (Fed. Cir. 1999). The term "impulse response" in **claims 1-25, 38-42, 44, and 45** is used by these claims (as best understood) to mean "a signal or signal component"; e.g., "said signal containing an input signal and multiple impulse responses, said multiple impulse

responses to be adaptively filtered" and "generating an estimate of an impulse response corresponding to each of said multiple impulse responses; generating a sum of said estimates; and generating an error signal representing the difference between said signal and said sum of said estimates" or equivalent in claims 1, 6, and 11; "generating estimates of impulse responses ... and generating an error signal representing the difference between a desired signal on said downstream channel and a sum of said estimates" in claim 18; "generating an estimate of an impulse response corresponding to echo paths ... and a difference circuit for generating an error signal representing the difference between a signal on said downstream channel representing sound at said second location and said estimate" in claim 38, while the accepted meaning (in digital signal processing) is "the response of a system or network to a unit sample sequence." It is unconventional to speak of a signal "comprising an impulse response" or of "subtracting a sum of estimates of impulse responses from a signal" as in claim 1, for instance. The term is indefinite because the specification does not clearly redefine the term. Moreover, Applicant contrarily uses the term "impulse response" in **claims 26-37 and 43** according to the conventional definition recited above: "developing an estimated impulse response corresponding to each of said upstream channels ... convolving each of said estimated impulse responses with a signal on the corresponding one of said upstream channels to generate an estimate corresponding to each of said upstream channels; and summing each of said individual estimates" in claims 26 and 33. Thus, one of ordinary skill in the art could not reasonably resolve the meaning and scope of the claims containing this term (i.e., **all the claims**). Since a single definition of the term

cannot be consistently applied to the claims, the examiner will apply either definition in the prior-art rejections that follow, as convenient.

11. **Claims 1, 6, 11, 18, and 26** use the singular and plural forms of the word "estimate" alternately and inconsistently (e.g., reciting an "estimate"; and then later reciting "said estimates", "said individual estimates"; and in some cases again later reciting "said estimate" [claims 6, 18, 26, and 33]). The inconsistent use of the plural and singular forms renders the claims indefinite because one of ordinary skill in the art could not reasonably ascertain the scope of the invention claimed.

12. Regarding **claims 4, 5, 9, 10, 14-17, 21-24, 29-32, 36, 37, 41-45, and 49-52**, the equations of claims **4, 5, 9, 10, 14, 15, 21, 22, 29, 30, 36, 37, 41, 42, 49, and 50** contain the symbols $\underline{\hat{\boldsymbol{h}}}$, m, λ_f, μ_u, $\boldsymbol{S}$ (or $\boldsymbol{s}$?), $\boldsymbol{S}_u/\boldsymbol{s}_u$, $\boldsymbol{D}^H$, $\underline{e}$, $\underline{\hat{y}}$, $\boldsymbol{G}$, and $\boldsymbol{D}$ that are not defined in the claims. At the time the present invention was made, it was known to employ these symbols to represent a wide variety of items (variables/parameters); and equations of the general form of those recited in the claims were common in the prior art (e.g., Equation 21 of Mansour and Gray); although these symbols may be given specific meanings in the examples of the specification, limitations (such as specific meanings of these symbols and the metes and bounds of the invention defined thereby) are not imported from the specification to the claims in the examination process. Also, Applicants apply a first set of meanings to the symbols $\underline{\boldsymbol{e}}$, $\boldsymbol{D}$, $\underline{\hat{\boldsymbol{h}}}$, $\boldsymbol{S}$, $\boldsymbol{s}$, $\boldsymbol{S}_u$, and $\boldsymbol{s}_u$ with respect to a single-channel AEC embodiment at pages 18-28 of the specification, then

apply a second, different set of meanings to the same symbols with respect to a multi-channel AEC embodiment at pages 28-32. Moreover, Applicants explicitly state at the conclusion of the specification at page 38, lines 6-8, "Accordingly, the foregoing description is by way of example only, and is not limiting. The invention is limited only as defined in the following claims and equivalents thereto." Finally, because of the apparent inconsistent use of font sizes and types in the equations of the specification and claims, it is difficult or impossible to distinguish some uppercase symbols from their lowercase counterparts or boldface symbols from normal-weight counterparts (e.g., the symbol "***s***" or "***S***" in these claims) according to the convention described at page 18, lines 13-17 of the specification (even if this exemplary convention were considered limiting). Thus, no limiting definition of the symbols employed in the claims is provided and, consequently, the claims are indefinite because one of ordinary skill in the art could not reasonably ascertain the metes and bounds of the claimed invention.

13. **Claim 11** recites the limitation "an estimate of an impulse response corresponding to each of said first and second impulse responses " in lines 11-13 of the claim. There is insufficient antecedent basis for these limitations in the claim. As demonstrated by Hirano et al., a single impulse response estimate can at once correspond to each of first and second impulse responses. There is no prior mention of plural estimates of an impulse response. Also, while the claim recites "multiple impulse

responses" at lines 3-4 of the claim, no prior mention is made of specific "first and second impulse responses".

14. **Claims 26 and 33** recite the limitations "said multiple upstream channels" in line 7 of the claims; "said corresponding upstream channel" in lines 8-9; "said downstream channel" in line 9; and "said upstream channels" in line 11. There is insufficient antecedent basis for these limitations in these claims. No prior mention is made of "multiple upstream channels", a "corresponding upstream channel", a "downstream channel", or "upstream channels".. Similarly, **claims 27, 28, 34, and 35** recite these limitations or variations thereof without proper antecedent basis in the claims.

15. **Claim 33** uses the singular and plural forms of the term "estimated impulse response" inconsistently (e.g., reciting an "estimated impulse response". The inconsistent use of the plural and singular forms renders the claims indefinite because one of ordinary skill in the art could not reasonably ascertain the scope of the invention claimed.

16. **Claims 43, 51, and 52** recite the limitation "said finite impulse response circuit" in lines 1-2 of claims 43 and 52 and lines 3-4 of claim 51. There is insufficient antecedent basis for this limitation in the claim. Although "finite impulse response filters" are previously recited, no prior mention is made of "finite impulse response circuits".

Claim Rejections - 35 USC § 102

17. The following is a quotation of the appropriate paragraphs of 35 U.S.C. 102 that form the basis for the rejections under this section made in this Office action:

> A person shall be entitled to a patent unless –
>
> (b) the invention was patented or described in a printed publication in this or a foreign country or in public use or on sale in this country, more than one year prior to the date of application for patent in the United States.

18. **Claims 1, 6, and 11** are rejected under 35 U.S.C. 102(b) as being anticipated by Hirano et al. (US 5,396,554).

19. Regarding **claim 11,** Hirano et al. disclose in Fig. 3 a multi-channel echo canceling apparatus (100) for transmitting a signal over a channel (26) in a multiple-channel communication apparatus where said signal includes an input signal (22) and multiple "impulse responses" (i.e., echoes 15 and 16) (The acoustic echo signals reproduced by loudspeakers 13 and 14, convolved with the impulse responses of acoustic echo paths 15 and 16, respectively, and then received by microphone 19 may be loosely described as "containing the impulse responses" of the acoustic echo paths, the apparent meaning intended by Applicant.), wherein said multiple impulse responses (echoes) are to be adaptively filtered (column 9, lines 24-27), said apparatus comprising:

a transmitter (inherently) for generating a data signal for transmission via a communication channel (26), wherein said signal includes an input signal (22) and

multiple impulse responses (acoustic echo signals 15 and 16) wherein said multiple impulse responses (echoes) are to be adaptively filtered (column 9, lines 24-27);

an adaptive filter circuit (103) for generating an estimate of an impulse response corresponding to each of said [multiple] impulse responses (column 9, lines 50-54);

a subtracter circuit (105) for generating an error signal (output signal 26) representing the difference between said data signal (24) and a sum of said estimates (adaptive filter 103 simultaneously generates a combined echo ["impulse response"] estimate signal that is a sum of estimates of the echo signals ["impulse responses"] contained in data signal 24 due to acoustic echo paths 15 and 16, based on the assumption that one of the echo signals is simply a delayed replica of the other, as described at column 4, lines 1-19);

wherein said estimates are generated using a frequency domain recursive least squares algorithm (Hirano et al. discloses at column 17, lines 41-52 that an RLS [recursive least squares] adaptive algorithm may be employed, and also, that the adaptive filter may operate in the frequency domain).

20. Regarding **claims 1 and 6**, in normal operation, the apparatus of Fig. 3 of Hirano et al. clearly performs the methods claimed, according to the description above regarding claim 11.

Claim Rejections - 35 USC § 103

21. The following is a quotation of 35 U.S.C. 103(a) which forms the basis for all obviousness rejections set forth in this Office action:

> (a) A patent may not be obtained though the invention is not identically disclosed or described as set forth in section 102 of this title, if the differences between the subject matter sought to be patented and the prior art are such that the subject matter as a whole would have been obvious at the time the invention was made to a person having ordinary skill in the art to which said subject matter pertains. Patentability shall not be negatived by the manner in which the invention was made.

22. Claims **18, 26, and 33** are rejected under 35 U.S.C. 103(a) as being unpatentable over Hirano et al. (US 5,396,554).

23. Regarding **claims 18, 26, and 33**, Hirano et al. disclose in Fig. 2 a prior-art system and associated inherent method of multi-channel communication (comprising cancelling acoustic echo distortion in a communication system) between at least first and second locations (as generally disclosed, e.g., column 1, lines 13-50), said method comprising the steps of:

transmitting multiple channels of information (501 and 502) upstream from said first location (not illustrated, but inherently present, providing received signals 501 and 502 and receiving transmission signals 516 and 517) to said second location (as illustrated);

transmitting at least one additional channel (516) of information downstream from said second location to said first location;

generating estimates (the impulse responses of adaptive filters 531 and 532) of impulse responses (developing an estimated impulse response) corresponding to distortion paths (corresponding to each of said [channels from the first location to the second location] 501 and 502, and that models an interference path at said second location from said corresponding [channel from the first location to the second location] to said [channel from said second location to said first location]) (acoustic echo paths 505 and 506) at said second location coupled between each of said multiple upstream channels [channels from the first location to the second location] (501 and 502) and said downstream channel [channel from said second location to said first location] (516) (convolving each of said estimated impulse responses with a signal on the corresponding one of said [channels from the first location to the second location] to generate an estimate corresponding [to] each of said [channels from the first location to the second location]; and summing the individual estimates, according to the alternate meaning applied to the term "impulse response" in claims 26 and 33); and

generating an error signal (the output of subtracter 539) representing the difference between a desired signal on said downstream channel and a sum of said estimates (535 and 536) (inherently comprising summing each of said individual estimates) and transmitting said error signal (516) to said first location.

Hirano et al. do not disclose that said estimate[s] [are] generated using a frequency domain recursive least squares algorithm in the prior-art system of Fig. 2; however, Hirano et al. disclose at column 17, lines 41-52 that an RLS [recursive least squares] adaptive algorithm may be employed, and also, that the adaptive filter of the

invention may operate in the frequency domain. At the time the present invention was made, the RLS algorithm was well known in the art to provide superior performance (faster convergence and better tracking) relative to the more commonly used LMS (least mean squares) algorithm in applications where the undesired signal is highly correlated with the desired signal (a condition recognized by Hirano et al. for the multi-channel echo cancelling arrangement – column 4, lines 6-11). (Since adaptive filter 531 in the prior-art arrangement should ideally only cancel the portion of the mixed signal 514 due to the echo of received signal 501, while allowing adaptive filter 532 to cancel the portion of mixed signal 514 due to the echo of received signal 502, as was well known in the art, the received signal 502 of the other channel is a "desired" signal with respect to the operation of adaptive filter 531.) Also, frequency-domain adaptive filtering techniques were known to provided more efficient implementations for adaptive filters having a large number of taps (as typically required for echo cancellation).

At the time the present invention was made, it would have been obvious to one of ordinary skill in the art to employ a frequency-domain recursive least squares algorithm in the multi-channel echo cancellation method of the prior art according to Fig. 2 of Hirano et al. as suggested by Hirano et al. for the system and method of that invention, in order to obtain the known benefits of improved performance and efficiency in cancelling the highly-correlated signals of the typical stereo teleconferencing system.

24. **Claims 2, 3, 5, 7, 8, 10, 12, 13, 15-17, 19, 20, 22, 25, 27, 28, 30, 31, 34, 35, 37, 39, 40, 42-45, 47, 48, and 50-52** are rejected under 35 U.S.C. 103(a) as being unpatentable over Hirano et al. in view of Mansour and Gray ("Unconstrained Frequency-Domain Adaptive Filter" [Reference V]).

25. Regarding **claims 2, 3, 7, 8, 12, and 13**, as described above, Hirano et al. disclose an apparatus and associated method of normal operation meeting the limitations of claims 1, 6, and 11. Hirano et al. do not disclose that the adaptive filter of the apparatus generates each of said estimates by diagonally decomposing by Fourier transformation a circulant matrix formed by augmentation of said input signal (or as more specifically claimed in claims 3, 8, and 13, forming a matrix of vectors representing said input signal; augmenting said matrix to form a circulant matrix; and decomposing said circulant matrix by Fourier transformation to form a diagonal matrix).

Mansour and Gray disclose generally an adaptive filter for use in applications such as echo cancellation (page 726, first paragraph) that generates an estimate of an impulse response in part by diagonally decomposing by Fourier transformation a circulant matrix ($\mathcal{X}_k$) formed by augmentation of an input signal. Mansour and Gray do not describe the formation of the circulant matrix $\mathcal{X}_k$ as comprising the separate steps of forming a matrix of vectors representing said input signal and augmenting the matrix to form a circulant matrix; rather the document implies a more direct formation of the circulant matrix $\mathcal{X}_k$ by forming a vector of length $2N$ of consecutive input samples and

creating a circulant matrix by placing that vector in the first row, then forming each consecutive row by rotating the row above to the right one position. The matrix $\mathcal{X}_k$ in Equation 7 of page 727 of Mansour and Gray can be resolved (by separating it into four *N* x *N* matrices) into an *N* x *N* matrix **X** (occurring twice in the augmented matrix) and a matrix **X'** (also occurring twice in the augmented matrix) equivalent to that described by Applicant at page 20 of the specification as follows:

$$\chi_k = \mathbf{C} = \begin{bmatrix} \mathbf{X'} & \mathbf{X} \\ \mathbf{X} & \mathbf{X'} \end{bmatrix}, where$$

$$\mathbf{X} = \begin{bmatrix} x(N) & x(N+1) & x(N+2) & \cdots & x(2N-1) \\ x(N-1) & x(N) & x(N+1) & \cdots & x(2N-2) \\ x(N-2) & x(N-1) & x(N) & \cdots & \cdots \\ \cdots & \cdots & \cdots & \cdots & x(N+1) \\ x(1) & x(2) & \cdots & x(N-1) & x(N) \end{bmatrix} and$$

$$\mathbf{X'} = \begin{bmatrix} x(0) & x(1) & x(3) & \cdots & x(N-1) \\ x(2N-1) & x(0) & x(1) & \cdots & x(N-2) \\ x(2N-2) & x(2N-3) & x(0) & \cdots & \cdots \\ \cdots & \cdots & \cdots & \cdots & x(1) \\ x(N+1) & x(N+2) & \cdots & x(2N-1) & x(0) \end{bmatrix}$$

Thus, matrix $\mathcal{X}_k$ of Mansour and Gray is equivalent to that claimed by Applicants; and Applicants have not shown any benefit to forming such a matrix by the separate steps claimed. Matrix ($\mathcal{X}_k$) is then diagonally decomposed by Fourier transformation to form the diagonal matrix $\mathbf{X}_k$ (which could be named "***D***" without the exercise of any

inventive process). Mansour and Gray also do not describe the frequency-domain adaptive filter as a "recursive least squares" filter; however, since the complex conjugate transpose (a.k.a. the Hermitian) of a diagonal matrix (i.e., "***D***") is equivalent to the complex conjugate of the matrix, Applicants' Equations 15 and 16 at page 25 of the specification are equivalent to the equations of claim 5, which claim must include all the limitations of claim 1, and therefore must define a frequency domain recursive least squares algorithm, Applicants admit at page 26, lines 2-3 of the specification with regard to Equations 15 and 16, that "This algorithm is exactly the unconstrained frequency-domain adaptive filter proposed by Mansour and Gray". Thus, to the same extent that the adaptive filter of Applicants' invention employs a frequency-domain recursive least squares algorithm, so does that of Mansour and Gray. Mansour and Gray disclose in the abstract on page 726 that for a large number of taps (as required in typical acoustic echo cancelling applications) the disclosed adaptive filter offers significant reduction in computational requirements, as well as fast convergence for highly correlated input signals (as recognized by Hirano et al. at column 4, lines 6-10).

At the time the present invention was made, it would have been obvious to one of ordinary skill in the art to employ the frequency-domain adaptive filter of Mansour and Gray in the multi-channel echo cancelling method and apparatus of Hirano et al. by providing circuits to perform each of the recited steps in order to obtain the benefits described in the abstract of Mansour and Gray.

26. Regarding **claims 5, 10, and 15**, Applicants admit at page 26, with regard to the algorithm represented by Equations 15 and 16 on page 25, "This algorithm is exactly the unconstrained frequency-domain adaptive filter proposed by Mansour and Gray [3]". When Equations 52 and 53 on page 731 of Mansour and Gray are substituted into Equation 21 on page 728 and equivalent symbols used in the equations of these claims are substituted into the equations of Mansour and Gray, the equivalence is apparent. The following table summarizes the equivalence of the symbols used by Applicants to those used by Mansour and Gray:

Quantity or Parameter Represented	Mansour & Gray	Applicants
Freq. domain filter weight vector	$\boldsymbol{W}$	$\underline{\hat{\boldsymbol{h}}}$
Next block index	k+1	m
Current block index	k	m-1
Normalized convergence factor for all frequencies	α	μ_u
Estimate of input signal energy in i-th frequency bin	z	$\boldsymbol{S}_u$
diag(DFT(1st row of input matrix))	$\mathbf{X}$	$\boldsymbol{D}$
Energy smoothing constant for all frequencies	β	$1-\lambda_f$
Forgetting factor	$1-\beta$	λ_f
Freq. domain error signal vector	$\boldsymbol{E}$	$\underline{\mathbf{e}}$
Freq. domain data windowing matrix	$\boldsymbol{H}$	$\mathbf{G}$
Time domain data windowing matrix	$\boldsymbol{h}$	$\boldsymbol{W}$
Freq. domain convergence constant diagonal matrix	μ	$\mu_u \boldsymbol{S}_u^{-1}/2$
DFT of desired output voltage	$\boldsymbol{D}$	$\underline{\boldsymbol{y}}$

Substituting the matrix equivalents of Equations 52 and 53 of Mansour and Gray into Equation 21 gives:

$$\mathbf{W}_{k+1} = \mathbf{W}_k + 2\alpha\mathbf{z}_k^{-1}\mathbf{X}_k^H\mathbf{E}_k, where$$

$$\mathbf{z}_k = (1-\beta)\mathbf{z}_{k-1} + \beta\mathbf{X}_k\mathbf{X}_k^*$$

Substituting Applicants' equivalent symbols from the table above into these equations gives:

$$\hat{\underline{\mathbf{h}}}(m) = \hat{\underline{\mathbf{h}}}(m-1) + \mu_u\mathbf{S}_u^{-1}(m)\mathbf{D}^H(m)\mathbf{e}(m), where$$

$$\mathbf{S}_u(m) = \lambda_f\mathbf{S}_u(m-1) + (1-\lambda_f)\mathbf{D}^H(m)\mathbf{D}(m)$$

Since the complex conjugate transpose (Hermitian) of a diagonal matrix is equivalent to the complex conjugate, the equations are seen to be equivalent (as best understood, in view of the indefiniteness of these claims as described in the 35 USC 112, second paragraph rejection of these claims, above).

27. Regarding **claims 19, 20, 27, 28, 34, and 35**, Hirano et al. do not disclose that the adaptive filter of the prior art apparatus and method of Fig. 2 generates each of said estimates by diagonally decomposing by Fourier transformation a circulant matrix

formed by augmentation of said input signal (or as more specifically claimed in claims 20, 28, and 35, by forming a matrix of vectors representing said input signal [on said upstream channel]; augmenting said matrix to form a circulant matrix; and decomposing said circulant matrix by Fourier transformation to form a diagonal matrix).

As described above regarding claims 2, 3, 7, 8, 12, and 13, Mansour and Gray disclose a frequency-domain adaptive filter for applications such as acoustic echo cancellation that for a large number of taps (as required in typical acoustic echo cancelling applications) offers significant reduction in computational requirements, as well as fast convergence for highly correlated input signals (as recognized by Hirano et al. at column 4, lines 6-10). As further described above in regard to claims 2, 3, 7, 8, 12, and 13, the adaptive filter and method of Mansour and Gray is equivalent to that claimed, and the claimed adaptive filter and method are an obvious variation of that of Mansour and Gray.

At the time the present invention was made, it would have been obvious to one of ordinary skill in the art to employ the frequency-domain adaptive filter of Mansour and Gray in the multi-channel echo cancelling method and apparatus of the prior art disclosed by Hirano et al. by providing circuits to perform each of the recited steps in order to obtain the benefits described in the abstract of Mansour and Gray.

28. Regarding **claims 22, 30, and 37**, as described above in regard to claims 19, 20, 27, 28, 34, and 35, it would have been obvious to employ the frequency-domain adaptive filter of Mansour and Gray in the prior-art multi-channel acoustic echo

canceller of Fig. 2 of Hirano et al. Also, as described above in regard to claims 5, 10, and 15, Applicants admit at page 26 of the specification that the claimed algorithm is "exactly the unconstrained frequency-domain adaptive filter proposed by Mansour and Gray"; and the examiner has demonstrated the equivalence above.

29. Regarding **claim 25**, in the apparatus and associated inherent method of operation of the prior-art echo canceller of Fig. 2 of Hirano et al., employed in a teleconferencing system as described at column 1, lines 18-36, the multiple channels of upstream information (501 and 502) comprise sound generated at the first location and the distortion paths comprise echo paths (505 and 506) at the second location coupled between each of said multiple upstream channels and said downstream channel (516).

30. Regarding **claim 31**, the multiple upstream channels in the apparatus and method of Fig. 2 of Hirano et al. comprise a first channel (501) and a second channel (502).

31. **Claims 23 and 24** are rejected under 35 U.S.C. 103(a) as being unpatentable over Hirano et al. in view of Mansour and Gray ("Unconstrained Frequency-Domain Adaptive Filter" [Reference V]) as applied to claim 18 above, and further in view of Benesty et al. ("A Better Understanding and an Improved Solution to the Problem of Stereophonic Acoustic Echo Cancellation" [Reference U]).

32. Regarding **claim 23**, neither Hirano et al. nor Mansour and Gray disclose introducing a non-linear transformation module into at least one of said upstream paths.

Benesty et al. (including Applicants) disclose in Reference U a method of improved stereophonic echo cancellation in which the problem of a high degree of correlation between the signals of the two "upstream" channels is partially addressed by placing a "non-linear transformation module" in each upstream signal path (page 305). Benesty et al. disclose at pages 305-306, sections 6 and 7 that the non-linear transformation improves the operation (reduces the degree of misalignment) of the stereophonic echo canceller.

At the time the present invention was made, it would have been obvious to one of ordinary skill in the art to employ the non-linear transformation module of Benesty et al. in the prior-art teleconferencing system of Fig. 2 of Hirano, employing the frequency-domain adaptive filter of Mansour and Gray as described above, in order to reduce the degree of correlation between the "upstream" channels and thus further improve the level of performance of the echo canceller.

33. Regarding **claim 24**, in the prior-art teleconferencing system of Fig. 2 of Hirano, employing the frequency-domain adaptive filter of Mansour and Gray and the non-linear transformation of Benesty et al. as described above, [generation of] each of said estimates comprises generating a model (the impulse responses of adaptive filter 531 or 532) of a distortion path (echo path 505 or 506) at said second location from said corresponding upstream channel (501 or 502) to said downstream channel (516) and

said step of generating said estimates further comprises the steps of convolving each of said estimates with a signal on the corresponding one of said upstream channels to generate an estimate for each individual one of said upstream channels (the inherent mode of operation of FIR filters, such as 531 and 532), and summing each of said individual estimates (539).

34. **Claims 38 and 46** are rejected under 35 U.S.C. 103(a) as being unpatentable over Hirano et al. (US 5,396,554) in view of Benesty et al. ("A Better Understanding and an Improved Solution to the Problem of Stereophonic Acoustic Echo Cancellation" [Reference U]).

35. Regarding **claims 38 and 46**, Hirano et al. disclose in Fig. 2 a prior-art multi-channel teleconferencing apparatus comprising:

at least first and second upstream electrical paths (501 and 502) between a first location (not illustrated, but inherently present in a teleconferencing system as disclosed at column 4, lines 1-6) and a second location (as illustrated in Fig. 2) for transmitting acoustic signals from said first location to said second location;

at least one downstream electrical path (516) between said second location and said first location for transmitting acoustic signals from said second location to said first location;

a finite impulse response filter (the combination of 531 and 532) coupled between said upstream paths (501 and 502) and said downstream path (516) for

generating an estimate of an impulse response corresponding to echo paths (505 and 506) at said second location coupled between said at least first and second upstream channels and said downstream channel; and

a difference circuit (105) for generating an error signal (26) representing the difference between a signal (24) on said downstream channel representing sound at said second location and said estimate (the output of adaptive filter 103).

Hirano et al. do not disclose at least one non-linear transformation module coupled within each of one or more of said upstream paths, nor that the estimate is generated in the prior-art echo canceller of Fig. 2 using a frequency domain recursive least squares algorithm.

Hirano et al. disclose at column 17, lines 41-52 that an RLS [recursive least squares] adaptive algorithm may be employed, and also, that the adaptive filter of the invention may operate in the frequency domain. At the time the present invention was made, the RLS algorithm was well known in the art to provide superior performance (faster convergence and better tracking) relative to the more commonly used LMS (least mean squares) algorithm in applications where the undesired signal is highly correlated with the desired signal (a condition recognized by Hirano et al. for the multi-channel echo cancelling arrangement – column 4, lines 6-11). (Since adaptive filter 531 in the prior-art arrangement should ideally only cancel the portion of the mixed signal 514 due to the echo of received signal 501, while allowing adaptive filter 532 to cancel the portion of mixed signal 514 due to the echo of received signal 502, as was well known in the art, the received signal 502 of the other channel is a "desired" signal with respect

to the operation of adaptive filter 531.) Also, frequency-domain adaptive filtering techniques were known to provided more efficient implementations for adaptive filters having a large number of taps (as typically required for echo cancellation).

Benesty et al. (including Applicants) disclose in Reference U a method of improved stereophonic echo cancellation in which the problem of a high degree of correlation between the signals of the two "upstream" channels is partially addressed by placing a "non-linear transformation module" in each upstream signal path (page 305). Benesty et al. disclose at pages 305-306, sections 6 and 7 that the non-linear transformation improves the operation (reduces the degree of misalignment) of the stereophonic echo canceller.

At the time the present invention was made, it would have been obvious to one of ordinary skill in the art to employ a frequency-domain recursive least squares algorithm in the multi-channel echo cancellation method of the prior art according to Fig. 2 of Hirano et al. as suggested by Hirano et al. for the system and method of that invention, in order to obtain the known benefits of improved performance and efficiency in cancelling the highly-correlated signals of the typical stereo teleconferencing system, and further to employ the non-linear transformation module of Benesty et al. in order to reduce the degree of correlation between the "upstream" channels and thus further improve the level of performance of the echo canceller.

36. **Claims 39, 40, 42-45, 47, 48, and 50-52** are rejected under 35 U.S.C. 103(a) as being unpatentable over Hirano et al. (US 5,396,554) in view of Benesty et al. ("A Better Understanding and an Improved Solution to the Problem of Stereophonic Acoustic Echo Cancellation" [Reference U]) as applied to claims 38 and 46 above, and further in view of Mansour and Gray (("Unconstrained Frequency-Domain Adaptive Filter" [Reference V]).

37. Regarding **claims 39, 40, 47, and 48**, Hirano et al. do not disclose that the adaptive filter of the prior art apparatus and method of Fig. 2 generates each of said estimates by diagonally decomposing by Fourier transformation a circulant matrix formed by augmentation of said input signal (or as more specifically claimed in claims 40 and 48, by forming a matrix of vectors representing said input signal [on said upstream channel]; augmenting said matrix to form a circulant matrix; and decomposing said circulant matrix by Fourier transformation to form a diagonal matrix).

As described above regarding claims 2, 3, 7, 8, 12, and 13, Mansour and Gray disclose a frequency-domain adaptive filter for applications such as acoustic echo cancellation that for a large number of taps (as required in typical acoustic echo cancelling applications) offers significant reduction in computational requirements, as well as fast convergence for highly correlated input signals (as recognized by Hirano et al. at column 4, lines 6-10). As further described above in regard to claims 2, 3, 7, 8, 12, and 13, the adaptive filter and method of Mansour and Gray is equivalent to that

claimed, and the claimed adaptive filter is an obvious variation of that of Mansour and Gray.

38. Regarding **claims 42 and 50**, as described above in regard to claims 39, 40, 47, and 48, it would have been obvious to employ the frequency-domain adaptive filter of Mansour and Gray in the prior-art multi-channel acoustic echo canceller of Fig. 2 of Hirano et al. Also, as described above in regard to claims 5, 10, and 15, Applicants admit at page 26 of the specification that the claimed algorithm is "exactly the unconstrained frequency-domain adaptive filter proposed by Mansour and Gray"; and the examiner has demonstrated the equivalence above.

39. Regarding **claim 43**, in the prior-art teleconferencing system of Fig. 2 of Hirano, employing the frequency-domain adaptive filter of Mansour and Gray and the non-linear transformation of Benesty et al. as described above, the finite impulse response filter circuit comprises a finite impulse response circuit (531 and 532) corresponding to each of said upstream channels (501 and 502) for generating an impulse response that models an impulse response corresponding to an echo path (505 or 506) at said second location from said corresponding upstream channel (501 or 502) to said downstream channel (516), each finite impulse response filter coupled between said corresponding upstream path and said downstream path (as shown in Fig. 2).

40. Regarding **claim 44**, the prior-art teleconferencing system of Fig. 2 of Hirano, employing the frequency-domain adaptive filter of Mansour and Gray and the non-linear transformation of Benesty et al. as described above comprises a summing circuit (539) for summing said estimates.

41. Regarding **claim 45**, the prior-art teleconferencing system of Fig. 2 of Hirano, employing the frequency-domain adaptive filter of Mansour and Gray and the non-linear transformation of Benesty et al. as described above comprises:

at least first and second microphones (not illustrated, but as first microphones 509 and 510 at the illustrated second location) at said first location for receiving sound, said microphones coupled to said first and second upstream electrical paths (not illustrated, but as 516 and 517, inherently coupled to illustrated upstream paths 501 and 502), respectively;

at least first and second speakers (503 and 504) at said second location coupled to said first and second upstream electrical paths (501 and 502), respectively, for recreating said sound from said first location at said second location;

at least a third microphone (509) at said second location for receiving sound, said microphone coupled to said downstream electrical path (516);

at least a third speaker (not illustrated, but as first speaker 503 at the illustrated second location) at said first location coupled to said downstream electrical path (516 at

second location to 501 at first location) for re-creating said sound from said second location.

42. Regarding **claim 51**, the prior-art teleconferencing system of Fig. 2 of Hirano, employing the frequency-domain adaptive filter of Mansour and Gray and the non-linear transformation of Benesty et al. as described above comprises a difference circuit (539) coupled to an output of the finite impulse response filter circuit (comprising filters 531 and 532) for coupling to said downstream path (516) for generating an error signal representing the difference between a signal on said downstream path representing sound at said second location and said estimate (as illustrated).

43. Regarding **claim 52**, the finite impulse response filter circuit of the prior-art teleconferencing system of Fig. 2 of Hirano, employing the frequency-domain adaptive filter of Mansour and Gray and the non-linear transformation of Benesty et al. as described above comprises multiple finite impulse response [filter] circuits for each of said upstream channels for each of said upstream channels for generating an impulse response that models an impulse response corresponding to an echo path at said second location from said correspond upstream channel to said downstream channel (as illustrated).

Conclusion

44. The prior art made of record and not relied upon is considered pertinent to applicant's disclosure.

45. Berthault et al. (US 6,738,480) discloses a frequency-domain stereo echo cancelling method and system similar to that disclosed by Applicants.

46. Shynk ("Frequency-Domain and Multirate Adaptive Filtering" [Reference W]) discloses a summary of a variety of frequency-domain adaptive filtering arrangements known at the time.

Any inquiry concerning this communication or earlier communications from the examiner should be directed to Tony M. Jacobson whose telephone number is 571-272-7521. The examiner can normally be reached on M-F 11:00-7:00.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Sinh N. Tran can be reached on 571-272-7564. The fax phone number for the organization where this application or proceeding is assigned is 703-872-9306.

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May 19, 2005

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